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The overall goal of this program of research is to specify the processes used by the auditory system to detect signals presented in noisy backgrounds. A wide variety of experimental approaches were used to examine these processes. The data suggest that subjects often detect the signal as a change in the spectral/temporal pattern of stimulus information. These results conflict with the classical models of simple auditory masking that suggest that subjects restrict their analysis to a narrow frequency band and a brief temporal window. Quantitative models of the process that compares information across spectral/temporal regions were developed, which combine "excitatory" and "inhibitory" components. While it has traditionally been assumed that quite different mechanisms govern monaural and binaural masking phenomena, very similar models were successfully applied to the two sets of data. Other significant results include a more complete description of internal noise processes, evidence that the external masker is not cancelled

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Abstract (Cont.)

by the binaural processor, empirical and theoretical evaluations of the efficiency of psychophysical procedures, and hardware and software developments to aid psycho-acoustic research. Overall, the work examined issues and models of contemporary interest and thus has implications for auditory theory in general and for the study of auditory pattern analysis and auditory masking in specific.

BINAURAL MASKING: AN ANALYSIS OF MODELS

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Binaural Masking: An Analysis of Models
Grant AFOSR 86-0298
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I. RESEARCH OBJECTIVES

The ultimate goal of the project is to specify the transformations of the auditory stimulus used by the subject to determine the presence or absence of a signal in an auditory masking task, with particular emphasis on the role of processes that compare information in the frequency domain and in the time domain, and on the relation between monaural and binaural effects. The study of auditory masking phenomena has far-reaching implications: 1) because, while all of auditory theory is based in whole or in part on the results of masking studies, the mechanisms underlying these phenomena are poorly understood; 2) because the noise reduction strategies used by the auditory system represent a basic form of auditory pattern analysis, which must be addressed when modeling more complex auditory processing; 3) because it is often critical to specify and to optimize human performance in noisy environments or through degraded communication channels; and 4) because the damaged ear has particular problems in noise, for which clinical solutions must be found.

Within our approach it is assumed that the behavior of the subject can be modeled by a system that on each trial computes a single "decision variable," which in the manner described by the Theory of Signal Detectability provides the basis of the subject's decision about the presence or absence of the signal. Within this framework the researcher's task is to specify this decision variable. For the tone-in-noise detection tasks we have been investigating, classical models would argue that the decision variable should correspond closely to the stimulus energy within a narrow frequency band centered around the signal (i.e., the critical band) and within a brief temporal window that contains the signal. In this project traditional psychophysical procedures were combined with "new" techniques ("Molecular psychophysics") in order to demonstrate that classical models of masking are oversimplifications, to develop models that provide a more accurate description of the responses of the subject, and to delineate the relations between the mechanisms underlying monaural masking and those underlying binaural masking. A further aim of this project has been to devleop and evaluate devices, software, and procedures to facilitate psychophysical research.

II. SUMMARY

A number of psychophysical studies have been performed to examine the phenomena of monaural and binaural masking. Both single-channel and multichannel models have been fit to the data from "molecular" psychophysical studies. The multiple detector models provide a better fit to the molecular data and, along with the results of a number of experiments employing traditional techniques, indicate that subjects determine the presence of the signal based on a comparison of information across different spectral/temporal regions of the

stimulus. These results are compatible with other findings in the current literature, suggest a more global analysis of the stimulus than has typically been assumed, and provide a quantitative description of these phenomena. Other significant results include a more complete description of internal noise processes, evidence that the external masker is not cancelled by the binaural processor, evidence that similar models can be used to predict monaural and binaural masking data, and evidence that remote masking and suppression may be related phenomena. In addition, adaptive staircase techniques have been examined, a software noise generator has been implemented, and a device to control the presentation of high quality sound in auditory experiments has been designed and implemented.

III. STATUS OF THE RESEARCH

Our research on monaural masking and on binaural masking proceeds in parallel, with considerable interdependence. Additional support for this research has been provided by grants from NSF (BNS-85-11768) "Analysis of models of auditory masking," period of support July 15, 1985 through July 14, 1988, R.H. Gilkey, PI, and (BNS-87-20305) "Analysis of models of auditory masking," period of support January 1, 1988 through June 30, 1989, R.H. Gilkey, PI.

Profile analysis in noise

In an experiment on diotic masking we investigated the effect of randomizing the overall level of the stimulus on the detectability of a relatively brief tonal signal, as a function of the duration and the bandwidth of the masking stimulus. If the subject bases his decision on the energy in a single critical band and a single temporal integration window, it should be possible to disrupt his performance by randomizing the overall level of the stimulus (thus randomizing the energy in that critical band and temporal integration window). The approach is based on the profile analysis experiments of Green [Am. Psychol. 38: 133-142, 1983]. Here, however, the background is random noise rather than a tone complex. When the masker is narrowband and short in duration, there is a substantial effect of randomizing overall level. However, when the masker is wideband and long in duration, the effect of randomizing overall level is negligible. Apparently, some information present in the spectral fringe (that portion of the masker that does not overlap with the signal in the frequency domain) and temporal fringe (that portion of the masker that does not overlap with the signal in the temporal domain) can be used to overcome the effects of randomizing overall level. When the bandwidth and the duration of the masker are manipulated separately, the results suggest that the information in the frequency domain is most important. These results call into question the classical critical band interpretation of auditory masking. Green [op. cit.] suggests that subjects in his profile analysis task base their decisions on an analysis of spectral shape. Our results suggest that a similar type of analysis may be occurring in tone-in-noise masking. Said differently, by comparing information in different spectral/temporal regions, the subject's decision variable can be scaled to be independent or nearly independent of overall level. The results were described in talks presented at CID and at the meeting of the AFOSR program in Auditory Pattern Recognition, and are included in Gilkey ["Spectral and temporal comparisons in auditory masking," in W.A. Yost and C.S. Watson (eds.), Auditory Processing of Complex Sounds, 26-36, 1987].

An experiment analogous to that of Gilkey [op. cit.] was conducted using the dichotic N0Sπ stimulus configuration. Again, the effect of randomizing overall level was much smaller under the wideband long-duration condition than under the narrowband short-duration condition. Again, when the effects of duration and bandwidth were investigated separately, the results suggest that the information in the spectral fringe is most important. The presence of an effect of randomizing overall level under the narrowband short-duration conditions is difficult to explain by simple models that are based solely on interaural differences. That is, even though the overall level is randomized, the interaural differences are not. Thus, in order to explain these data a model would, at a minimum, have to postulate the presence of multiplicative internal noise. On the other hand, the fact that the effect of randomizing level is eliminated when the bandwidth of the masker is wide suggests that the classical critical band interpretation of auditory masking is inadequate and that some process compares information in different spectral/temporal regions. These results were presented at the meeting of the AFOSR program in Auditory Pattern Recognition.

Binaural temporal masking

If the interaural phase of a noise masker is switched during the observation interval from in phase (N0) to 180° out of phase (Nπ) or from Nπ to N0, a brief interaurally out-of-phase signal (Sπ) will be about 15 dB more detectable in the N0 portion of the noise than in the Nπ portion. By investigating the change in detectability as a function of the delay (Δt) between the onset of the signal and the phase transition in the noise, the temporal response of the binaural system can be evaluated. The results of this case can be contrasted with a set of conditions in which the interaural phase of the noise is held constant (Nπ), but the level of the noise is reduced or increased by 15 dB halfway through the observation interval. Within a model such as the E-C model [N.I. Durlach, "Binaural signal detection: Equalization and cancellation theory" in J.V. Tobias (ed.), Foundations of Modern Auditory Theory II, 371-462, 1972] the first case produces a change of level only in the binaural channel. The second case produces a change in the level in the monaural channel as well. The curves that describe the relation between threshold and Δt can be thought of as temporal masking functions. They show, like traditional temporal masking data, that the decay of backward masking (cases where the N0 segment of the noise precedes an Nπ segment or where the lower intensity segment of the noise precedes the higher intensity segment) is more rapid than for forward masking. Double-sided exponential integration windows were fit to the forward and backward masking functions. The equivalent rectangular duration of the best-fitting window under monaural conditions ranged from 12-26 ms, somewhat larger than those estimated by Moore et al. [J. Acoust. Soc. Am. 83: 1102-1116, 1988]. The equivalent rectangular durations for the binaural conditions ranged from 41-83 ms, in the range estimated by Grantham and Wightman [J. Acoust. Soc. Am. 65: 1509-1517, 1979]. The observed differences between monaural and binaural conditions were taken as additional evidence that the binaural system responds sluggishly to changing stimulation [Grantham and Wightman, J. Acoust. Soc. Am. 63: 511-523, 1978]. A paper has been submitted [Kollmeier and Gilkey, 1989].

Effects of forward masker fringe

In studying the effects of a forward masker fringe, Yost [J. Acoust. Soc. Am. 78: 901-907, 1985] found that the threshold for a brief Sπ signal masked by a brief N0 masking noise was not changed when an Nπ forward masker fringe was added. This result was somewhat surprising in light of results such as those of McFadden [J. Acoust. Soc. Am. 40: 1414-1419, 1966] who showed that an N0 forward fringe substantially improved performance in an N0Sπ detection task, and concluded that the system uses the forward fringe as a diotic reference against which to detect the dichotic signal. If an N0 forward fringe provides a useful reference, it might be expected that an Nπ forward fringe would provide a detrimental reference. Yost's results also seemed to conflict with the interpretations of Kollmeier and Gilkey [op. cit.], who thought of the Nπ fringe as a forward masker. They found a gradual decrease in the amount of masking as a function of Δt (the delay between the onset of the signal and the phase transition in the noise), with a substantial effect of the Nπ segment of the noise even when the signal was presented well within the N0 segment of the noise. One possibility was that the function that relates threshold to Δt for the Nπ forward fringe condition intersects with the function that relates threshold to Δt for the pulsed masker condition at $\Delta t = 0$, even though the functions are different elsewhere. To resolve these questions, the detectability of an Sπ tonal signal was investigated as a function of Δt, in the presence of an N0 "masker" that was preceded by quiet, or by an Nπ "forward fringe" and followed by quiet or by an N0 or Nπ "backward fringe." The results show that the two functions were indeed different and that they did not intersect. Overall, the results failed to replicate those of Yost, showing instead that the presence of the Nπ forward fringe reduced detectability for all subjects under a wide variety of conditions. The results are a further indication that the auditory system uses information that does not overlap with the signal in the temporal domain. These data were presented to the MLD Society and to the Acoustical Society of America [Simpson and Gilkey, J. Acoust. Soc. Am. 82: S108(A), 1987]. A manuscript has been submitted [Gilkey, Simpson, and Weisenberger, 1988].

The interrelation between remote masking and suppression

We are partially replicating the experiments of Wegel and Lane [Physiol. Rev. 23: 226-285, 1924] on remote masking and of Duifhuis [J. Acoust. Soc. Am. 67: 914-927, 1980] on suppression, using modern adaptive psychophysical techniques; the same subjects participate in both experiments. In the remote masking task, the masker is a 602-Hz sinusoid and the signal is a simultaneous 1500-Hz sinusoid. In the suppression experiment the suppressor is a 602-Hz sinusoid, the masker is a 1500-Hz sinusoid, and the signal is a brief 1500-Hz sinusoid immediately following the masker. The results replicate those of the early studies. A preliminary analysis suggests that the nonlinear growth of remote masking can be seen as having two components, one suppressive and one excitatory. All of the results appear to be compatible with the prediction of the MBPNL model [Goldstein, "Updating cochlear driven models of auditory perception: A model for nonlinear auditory frequency analyzing filters," in H. Bouma and B. Elsendorn (eds.), Working models of human perception, 19-57, 1988; "Modeling two factor cochlear responses," presented at "Basic research in a clinical environment," July 5-7, Dedham, MA, 1989].

Molecular psychophysical analyses of models of masking

Monaural studies. In most studies of auditory masking, including those described above, both the stimulus and the performance of the subjects are described by their statistical properties (e.g., the average power in the stimulus and the average probability of a correct response). The outputs of models are described by their distributional properties and the average performance of a model is fit to the average performance of a subject. Another approach was described by Green [Psychol. Rev. 71: 392-407, 1964] and referred to as "molecular" psychophysics. In this approach, reproducible noise is used as a masker, such that the stimulus can be specified exactly on every trial. Similarly, the responses of the subject are considered on a trial-by-trial basis. The outputs of models are determined for each stimulus and the fit of the model is evaluated by comparing these outputs to the associated responses of the subjects.

Gilkey and Robinson [J. Acoust. Soc. Am. 79, 1499-1510, 1986] used computer models to predict subjects' responses to the individual noise samples of Gilkey, Robinson, and Hanna [J. Acoust. Soc. Am. 78: 1207-1219, 1985]. The parameters of the models were manipulated until the outputs were best able to predict the subjects' responses. The combination of a 50-Hz-wide filter, followed by a half-wave rectifier and an integrator with a 100-200-ms decay constant, predicted their responses relatively well. However, a model that formed a weighted combination of the outputs of several detectors, each of which processed information in a different spectral region, yielded even better predictions and suggested that subjects compare the spectrum near the signal frequency to other areas of the spectrum. A similar model that processed information over different temporal intervals suggests that subjects also compare the waveform during the signal interval to the waveform immediately before the onset of the signal.

Gilkey and Robinson [op. cit.] investigated a fairly small set of reproducible noise samples (25 noise-alone and 100 signal-plus-noise samples). We have recently replicated and extended their finding using a larger set of reproducible noise samples (150 noise-alone and 150 signal-plus-noise samples). Again, we began with a simple model composed of a single-tuned filter centered at the signal frequency, followed by a half-wave rectifier and an integrator with a leak. We sampled the output of the integrator at the end of the signal interval as the decision variable of the model. Our procedures had greater precision than those of Gilkey and Robinson and yield best-fitting bandwidths that were somewhat smaller than theirs (in the range of 26-49 Hz across subjects) and best-fitting decay constants that were somewhat shorter (in the range of 39-100 ms across subjects). We also investigated alternate or additional cues that the auditory system might be using to determine the presence of the signal. Specifically, we have considered cues related to the regularity of the envelope and the regularity of the fine structure of the waveform at the output of the model's initial filter. Preliminary results, however, suggest that these cues will not add greatly to the proportion of predicted variance. We also considered multi-channel models that combine the output of several detectors that process information in different spectral regions. The obtained spectral weighting functions, which describe how the model weights information across frequency, were quite similar to those of Gilkey and Robinson and suggest that subjects compare information in different spectral regions. A significant

advancement has been the description of these weighting functions with a relatively simple equation, which can be interpreted as the difference between "excitatory" and "inhibitory" Gaussian-shaped weighting functions. We also examined models that combine the output of a single filter over several brief temporal windows. The obtained temporal weighting functions were quite similar to those of Gilkey and Robinson, and suggest that subjects compare information immediately before the signal onset to information during the signal interval. These results strongly question classical critical band theory. Even though on average the subject's performance is unaffected by information that is outside a single critical band, when the responses to individual reproducible noise samples are investigated, it can be seen that the responses are dependent on the pattern of spectral information across a frequency range that is much wider than a single critical band. That is, subjects are more likely to report the presence of the signal if the stimulus spectrum is "peaked" near the signal frequency, independent of the overall height of the spectrum. A talk describing some of these results was presented to the Acoustical Society of America [Gilkey and Meyer, J. Acoust. Soc. Am. 82: S92(A), 1987] and at the meeting of the AFOSR program in Auditory Pattern Recognition. A manuscript is in preparation [Meyer and Gilkey, 1989].

If the spectral weighting functions derived by Gilkey and Robinson [op. cit.] and Gilkey and Meyer [op. cit.] provide a correct view of monaural processing, then it should be possible to add and subtract energy from different regions of the spectrum and change the observed value of P(y). 10 NA and 10 SN waveforms were selected from those investigated by Gilkey and Meyer. The energy in each of the waveforms was raised or lowered by 6 dB in each of seven 47-Hz-wide bands, centered from one octave below the signal frequency to one octave above the signal frequency, yielding 280 "modulated" waveforms. A different waveform was presented on each trial of a block, and a substantial portion of these waveforms were unmodulated. Although changes in P(y) with decrements are fairly small in general, the pattern of results is that which would be expected based on our previous weighting functions. The pattern for increments is somewhat unexpected. Although the predicted "inhibitory" effects are observed at low frequencies, the "excitatory" effect is broader than anticipated, and little evidence of inhibitory effects is observed at high frequencies. This apparent conflict can potentially be explained by assuming that the subject looks for a peak in the spectrum anywhere near the 500-Hz signal frequency. Under normal circumstances a peak, if present, would probably be at the signal frequency. However, under the modulated noise conditions peaks can occur over a wide range. This strategy would be equivalent to monitoring the output of several weighting-function models like those of Gilkey and Meyer, each tuned to different frequency regions, and choosing the largest output to use as a decision variable. We are presently investigating this possibility. The results were presented to the Acoustical Society of America [Gilkey et al., J. Acoust. Soc. Am. 84: S140(A), 1988b]. A manuscript is planned.

Binaural studies. We have also been using the molecular psychophysical approach to examine models of binaural hearing. The method is comparable to that in the monaural experiments except that the signal is presented 180° out of phase interaurally while the noise remains diotic. Initially, we have been examining a relatively small set of samples (25 noise-alone and 50 signal-plus-noise samples). Computing the output of a binaural model for a specific noise sample is not as straightforward as with a monaural model, because in

most binaural models the internal noise must be added at an earlier stage and cannot be thought of as random variation at the input to the decision stage. We have implemented a computer model similar to the E-C model [Durlach, op. cit.]. The model consists of two initial band-pass filters, one for each ear. The filtered waveform in the left channel is subject to random delay and multiplied by a random gain factor (i.e., "the internal noise"). The waveforms in each channel are subjected to fixed delays (equalization), and then the two channels are subtracted (cancellation). The differenced waveforms are then half-wave rectified and integrated to obtain the value of the model's decision variable. Because it is impractical to use Monte Carlo methods, the effect of the internal noise is incorporated by computing the decision variable for each of 100 equally probable combinations of random time delay and random gain factor from normal distributions of time delay and gain. One approach has been to investigate the relationship between the internal noise parameters of the model (i.e., the random time delay and gain factor) and the ratio of external to internal noise standard deviations (R) as described by Gilkey, Hanna, and Robinson [J. Acoust. Soc. Am. 69: S23(A), 1981] and others. Surprisingly, the value of R for the model is not a monotonic function of the magnitude of the model's internal noise parameters. Further, there are rather limited combinations of internal noise parameters that yield values of R comparable to those found with human observers.

A second approach has been to compute the average value of the model's decision variable for each waveform. This average value is used to predict the response of the subject on a waveform-by-waveform basis. The bandwidth of the initial filters of model, the standard deviation of a normal distribution of random delays, and the standard deviation of a normal distribution of random gains are varied in order to produce the best fit. Bandwidths fall in the range of 21 to 51 Hz, similar to the range observed for the N0S0 case. Standard deviations of delay distribution are between 119 and 200 μs and correspond roughly to the values estimated by Colburn and Durlach ["Models of binaural interaction," in E.C. Carterette and M.P. Friedman (eds.), Handbook of Perception, 467-518, 1978]. For two of three subjects, the values of the gain factor correspond relatively well to the estimates of Colburn and Durlach [op. cit.]. For the third subject, however, the gain factor is near zero, a somewhat anomalous result. With these parameter values, between 53 and 60% of the variance in the data of the subjects can be predicted, comparable to the single-channel model for the diotic case.

Next, a linear combination of the average output of the seven E-C elements was formed, each tuned to a different frequency regions from 350 to 650 Hz. The bandwidth of the initial filter, the standard deviation of the delay distribution, and the standard deviation of the gain distribution in each channel were set to the best-fitting values estimated for the single-channel model. We derived weighting functions described by the difference between excitatory and inhibitory Gaussian weighting functions. The resulting spectral weighting functions are quite similar in form to those for the diotic case and also produce a comparable increase in the proportion of predicted variance. A linear combination of the output of a single E-C element over seven different 21-ms subintervals of the signal interval also increases the proportion of predicted variance. However, the shapes of these temporal weighting functions are more inconsistent across subjects than was the case for the diotic condition and do not have an easily interpretable form. These molecular studies of binaural

masking were presented at the meeting of the AFOSR Program on Auditory Pattern Recognition and a manuscript is planned.

Estimates of internal noise as a function of signal frequency. As mentioned above, the internal noise parameters of a binaural model are of critical importance. The relation between R (the ratio of internal to external noise standard deviations) and signal frequency is being investigated under both N0S0 and N0Sπ conditions. The E-C model [Durlach, op. cit.] suggests that the influence of the random time delay should increase with signal frequency. Our estimates at 1500 Hz show values of R that are in general slightly lower than those at 500 Hz. Interpretation of these results is complicated by the fact, noted earlier, that the relation between the internal noise parameters of the E-C model and R is not simple.

Overall, these "molecular" results indicate that classical critical band theory provides an oversimplified view of processing in auditory masking tasks. The weighting functions provide a quantitative description of the way the system compares information across frequency and across time. Even though diotic and dichotic masking have typically been assumed to be governed by quite different mechanisms, similar models can be used to predict the responses of the subjects in both cases, yielding similar results.

Adaptive staircase techniques in psychoacoustics: A comparison of human data and a mathematical model

We compared two common adaptive staircase rules, the "one up-two down" rule and the "one up-three down" rule [Levitt, J. Acoust. Soc. Am. 49: 467-477, 1971] in combination with a two-alternative forced-choice procedure and with a three-alternative forced-choice procedure. The adaptive staircase tracks were modeled as Markov chains. The model predicts that threshold estimates obtained with the adaptive techniques should be equal to those derived with equivalent "fixed signal level" techniques. However, the human data indicate that the adaptive techniques tend to yield lower thresholds. The model predicts that the standard error of a threshold estimate obtained from an adaptive technique will decrease and approach zero as the number of trials used to compute the estimate increases. The human data show greater variability than predicted and approach a nonzero value as the number of trials increases. The predictions of the model suggest that the commonly used combination of the 2AFC procedure and the "1 up 2 down" rule is the least efficient method of estimating a threshold and that the 3AFC procedure in combination with the "1 up 3 down" rule is the most efficient method. The human data are less consistent, but generally show the combination of the 2AFC procedure and the "1 up 2 down" rule to be one of the least efficient methods. A manuscript has been published [Kollmeier, Gilkey, and Sieben, J. Acoust. Soc. Am. 83: 1852-1862, 1988].

Laboratory development

As described in our original proposal, a major part of our effort during the first period of the grant (July 15, 1986 to March 15, 1988) was spent upgrading and developing our computer facilities for laboratory control and data analysis. In October 1986 we installed a

new multiuser MicroVAX II for data analysis, graphics, and signal processing to support our monaural and binaural modeling efforts. In January 1987 we replaced the aging Nova 4x computer system, which had been used for experiment control, but had become increasingly unreliable. An existent SMS 1000 PDP11/73 computer was combined with newly developed hardware and software for stimulus presentation, response collection, timing, and device control. Programs were written to control specific experiments on auditory and tactile perception, allowing rapid access to large sets of waveforms, and the ability to present four independent waveforms with 16-bit accuracy at sample rates up to 40 kHz.

The MicroVAX II has 13 megabytes of main memory, a 71-megabyte (formatted) Winchester disk and a second 300-megabyte (formatted) Winchester disk, a 95-megabyte streaming tape drive, and nine serial ports. The following devices are connected to the system: an LA210 draft printer, two Hewlett-Packard HP7475A pen plotters, a Courier 2400-baud modem, and terminals, including VT330, VT240 and HP2623A graphics terminals, and VT220 display terminals. One port is reserved for communication with the PDP11/73 computer that is used for experiment control. Principal applications of the MicroVAX II are program development, data analysis, graphics, and modeling. Fitting algorithms, signal processing subroutines, graphics software, and modeling programs have been implemented, allowing us to analyze our experimental results. Ethernet hardware and software installed on the MicroVAX II allow high-speed communication between the MicroVAX II and other computers of the Research Department, including the Speech Perception Laboratory MicroVAX II that allows access to ILS signal-processing software, and a Research Department MicroVAX II system that allows access to word processing and spreadsheet software, from terminals in the Signal Detection Laboratory. For word processing we have purchased an LN03-plus laser printer. Also available on the network are four 8-port terminal servers. In addition, a microwave link between CID and Washington University permits communication between the Signal Detection Laboratory and the Washington University network of computers on the main campus and medical school campus. Presently, there are over 100 computers on the network, providing access to a variety of applications. These include library search and catalog facilities, national and international mail services such as BITNET and ARPANET, and signal-processing and statistical packages.

The Scientific Micro Systems SMS-1000 Model 40 consists of a PDP11/73 processor, 4 megabytes of main memory, an 85-megabyte Winchester disk, 1.2 megabyte floppy disk drive, six serial ports, and a real-time clock. Peripherals include a VT220 system console terminal and an LA100 draft printer. Two Micro Technology Unlimited Digisound 16-bit digital-to-analog and analog-to-digital conversion subsystems provide a total of four channels of D/A and four channels of A/D, allowing the presentation of independent waveforms to a maximum of four listeners. A parallel I/O interface permits communication with subject response boxes and control of programmable attenuators and electronic switches. The SMS-1000 is used for real-time experiment control and data acquisition for the auditory and tactile experiments conducted in the Signal Detection Laboratory.

Direct memory access control of the Micro Technology Unlimited DigiSound-16 with a Q-bus based computer

High-quality digital generation and recording of sounds is essential for many of our experiments. However, few high quality (16-bit) digital-to-analog and analog-to-digital subsystems were available for the PDP11/73 (Q-bus-based) computer, and most of these were expensive. After examining available options, we decided on the Micro Technology Unlimited (MTU) Digisound-16. However, in order to use this system we had to overcome four problems. First, the two "stereo" channels within a single Digisound are strobed with a 10 μs interchannel delay, producing a detectable interaural difference. Second, while the Digisound-16 has an 8-bit data path, the PDP11/73 has a 16-bit data path. Third, the Digisound-16 requires the data for the two stereo channels to be interleaved in its own buffer memory, while it is typically most convenient to store the waveforms in separate arrays in computer memory. Fourth, no specific mechanism was available to allow direct memory access (DMA) control of the Digisound 16. To overcome these problems we designed an interface that would allow as many as two DRV11-WA DMA interfaces to be connected to as many as two Digisound-16s. This interface design is now in use on three computer systems here at CID and benefits a number of other projects, including AFOSR grant #86-0335 "Auditory-acoustic basis of consonant perception," J.D. Miller, PI. MTU has used this design to produce a commercially-available device, which we hope will be of benefit to other auditory scientists. A manuscript describing the design of this interface is in preparation [*Gilkey and Partridge, 1989*].

Software generation of reproducible noise

A software shift-register noise generator has been implemented to generate reproducible noise for our experiments. Given three input values, this program will generate an arbitrary length (up to 5.2 days) reproducible two-state binary noise, which, when filtered, is approximately white and Gaussian. A paper describing this program and some of the properties of the noise it produces has been published [Gilkey, Robinson, and Frank, J. Acoust. Soc. Am. 83: 829-831, 1988].

IV. PUBLICATION ACTIVITY

Publications

Gilkey, R.H. (1987). "Spectral and temporal comparisons in auditory masking," in The Auditory Processing of Complex Sounds, edited by W.A. Yost and C.S. Watson (Erlbaum, New York), pp. 26-36.

Gilkey, R.H., Robinson, D.E., and Frank, A.S. (1988). "A software pseudorandom noise generator," J. Acoust. Soc. Am. 83, 829-831.

Kollmeier, B., Gilkey, R.H., and Sieben, U. (1988). "Adaptive staircase methods in psychophysics: An analysis using Markov theory," J. Acoust. Soc. Am. 83, 1852-1853.

Papers submitted

Gilkey, R.H., Simpson, B.D., and Weisenberger, J.M. "Effects of masker fringe on binaural detection," submitted to J. Acoust. Soc. Am.

Kollmeier, B., and Gilkey, R.H. "Binaural temporal masking: Evidence for sluggishness in binaural detection," submitted to J. Acoust. Soc. Am.

Papers in preparation

Gilkey, R.H., and Partridge, M.E. "Direct-memory-access control of the Micro Technology Unlimited Digisound-16 with a Q-bus-based computer," for submission to Behavior Res. Methods, Instrumentation and Computers.

Meyer, T.A., and Gilkey, R.H. "Modeling subject responses in a reproducible noise masking task," for submission to J. Acoust. Soc. Am.

Planned papers

Gilkey, R.H. "Models of binaural masking: A molecular psychophysical approach," planned for submission to J. Acoust. Soc. Am.

Gilkey, R.H., Simpson, B.D., and Hammoud, A. "Effects of manipulating the spectral shape of reproducible noise samples on the detection responses of human subjects," planned for submission to J. Acoust. Soc. Am.

V. PARTICIPATING PROFESSIONALS

Robert H. Gilkey

De Anza College, Cupertino, CA			
University of California, Berkeley, CA	B.A.	1976	Psychology
Indiana University, Bloomington, IN	Ph.D.	1981	Psychology

Dissertation title: "Molecular psychophysics and models of auditory signal detectability."

Wm. Michael Mudrovic

University of Missouri, St. Louis, MO	B.A.	1965	Spanish
University of Kansas, Lawrence, KS	Ph.D.	1976	Spanish
Washington University, St. Louis, MO	M.S.	1981	Speech and Hearing

VI. INTERACTIONS

Invited papers and conference talks

Gilkey, R.H. (1986). "Estimates of internal noise in binaural masking," MLD Society, Chicago, IL.

Gilkey, R.H. (1986). "Binaural processing," Central Institute for the Deaf, St. Louis, MO.

Gilkey, R.H. (1986). "Spectral and temporal comparisons in auditory masking," Central Institute for the Deaf, St. Louis, MO.

Gilkey, R.H. (1987). "Temporal interactions in binaural masking: Effects of forward masker fringe," MLD Society, Madison, WI.

Gilkey, R.H., and Meyer, T.A. (1987). "Modeling subject responses in a reproducible noise masking task," J. Acoust. Soc. Am. 82, S92-93(A).

Simpson, B.D., and Gilkey, R.H. (1987). "Effects of forward masker fringe on binaural detection," J. Acoust. Soc. Am. 82, S108(A).

Gilkey, R.H. (1987). "Binaural masking: An analysis of models," Meeting of the AFOSR Program in Auditory Pattern Recognition, Evanston, IL.

Gilkey, R.H. (1988). "Relations between monaural and binaural masking," Psychoacoustics Laboratory, Department of Psychology, University of Florida, Gainesville, FL.

Gilkey, R.H., Simpson, B.D., and Hammoud, A. (1988). "Effects of manipulating the spectral shape of reproducible noise samples on the detection responses of human subjects," J. Acoust. Soc. Am. 84, S140-141(A).